1

Signal processing

This invention relates to the processing of information signals such as the coding of audio signals or the embedding of watermarks in an information signal. .

5   Within the field of signal processing of information signals, such as audio signals, video signals or the like, it often is a problem to achieve a suitable trade-off between the desired effect of that processing and the general aim of not introducing perceivable distortions into the signal. Examples of signal processing include the coding of information signals and the embedding of watermarks in information signals.

10   Within the field of audio coding, an efficient encoding of an audio signal is desired without unduly degrading the perceptual quality of the audio signal when reconstructed from the encoded signal.

          Within the field of watermarking it is a general problem of controlling the tradeoff between the robustness of an embedded watermark and its transparency: On the one

15   hand, it is desirable to obtain an embedded watermark which is robust against signal processing, such as coding, compression, etc. On the other hand, the embedded watermark should not modify the host signal in a way that causes perceivable distortions, e.g. audible or visible distortions. Hence, the perceptual quality of the information signal should not be affected.

20   In particular, many watermarking schemes for audio signals use masking models of the human auditory system where a masking threshold is determined corresponding to a maximum allowable embedding strength that still ensures an inaudible embedding. Hence, the amount of watermark energy to be added to a host audio signal is determined by the amount of noise that can be masked by the signal. This headroom is

25   generally referred to as the mask-to-noise ratio.

          Most watermarking systems use a certain embedding model to determine the mask-to-noise ratio and to control the watermarking generation or the mixing of the watermark into the host signal. One may describe this as a two stage process: a) determine the headroom available for embedding a watermark and b) derive and apply a certain cost

2

function to the watermark generation and mixing stages. Both of these stages are part of the so-called watermark embedding model.

US patent no. 6,345,100 discloses a method of embedding a watermark in an audio signal. The method comprises determining a basis signal from the audio signal the inclusion of which would be imperceptible by a human listener. A watermark is embedded into the basis signal and the resulting watermark signal is mixed into the audio signal.

A shortcoming of such prior art approaches is that, due to model approximations and imperfections, they do not maintain a constant perceptual quality of the information signal for a given embedding model.

It is an object of the present invention to improve the perceptual quality of the processed information signal.

The above and other problems are solved by a method of processing an information signal, the method comprising:
- applying a signal modification process to an information signal resulting in a processed signal, said signal modification process being controlled by at least one control parameter;
- comparing the processed signal with the information signal to determine a measure of perceptual quality of the processed signal;
- adjusting said at least one control parameter in response to the determined measure of perceptual quality.

By determining a measure of perceptual quality of the processed signal and adjusting the parameters controlling the signal modification process, the perceptual quality of the processed signal is improved.

It is an advantage of the invention that it provides an automatic control of the parameters controlling the signal processing, thereby providing a uniform perceptual quality of the processed signal.

The term information signal comprises any signal representing perceivable information which may be perceivably distorted by the signal processing. Examples of such signals include audio signals, video signals, multimedia signals, signals representing images, or the like. The information signal may be embodied as an analog signal or a digital signal.

The term signal modification process comprises any process which modifies the information signal and which may cause perceivable distortions of the information

3

content represented by the information signal. Such a signal modification process may comprise the encoding of a signal, the combination of the information signal with other signals, such as embedding of information, e.g. a watermark, etc. The modification process may comprise one or more sub-processes, such as segmenting, Fourier transformation or

5   other signal transformations, filtering, mixing, etc.

The measure of perceptual quality may include any suitable quantity, parameter, or the like that is indicative of a perceivable change in the information content of the information signal caused by the signal modification process. Examples of such measures include a suitable cost function quantifying the artifacts of the signal processing in question.

10   An example of a method for measuring the perceptual quality of audio signals is described in Thilo Thiede et al., "PEAQ - The ITU Standard for Objective Measurement of Perceived Quality", J. Audio Eng. Soc., Vol. 48, No. 1/2, 2000.

It is a further advantage of the invention that the automatic control is based on a perceptual measure, thereby taking into account the artifacts of the signal processing that

15   influence the perceptual quality. Examples of such artifacts include a variation in bandwidth, noise loudness asymmetry, temporal masking behavior, or the like.

Hence, it is an advantage that artifacts may be considered other than those considered by the actual signal processing.

The step of comparing the processed signal with the information signal may

20   comprise one or more processing steps of one or both of the signals, such as filtering steps, signal transformations, e.g. Fourier transformations, decoding steps, etc. Furthermore, the step of comparing may comprise any suitable processing steps for determining the measure of perceptual quality.

The adjusting of the control parameters may comprise any suitable control

25   mechanism, e.g. a control loop comparing the measure of perceptual quality with a corresponding target value and correspondingly adjusting the control parameter, or the like. The adjustment may comprise an increase of the parameter, a decrease of the parameter, or the adjustment may comprise leaving the parameter unchanged.

Examples of control parameters include the embedding strength of a

30   watermark signal, the embedding frame size in a frame-based embedding process of a watermark signal, a frequency/time allocation, a parameter controlling the bit allocation of a quantizer in an audio coder, etc.

In a preferred embodiment, the method further comprises dividing the information signal into a sequence of segments of the information signal;

4

wherein the step of applying the signal modification process comprises applying the signal modification process to a first one of said segments of the information signal resulting in a first segment of the processed signal;

wherein the step of comparing the processed signal with the information signal comprises comparing said first segment of the processed signal with said first segment of the information signal;

and wherein the method further comprises applying at least a part of the signal modification process to a second one of said segments of the information signal resulting in a second segment of the processed signal, the at least part of the signal modification process being controlled by said adjusted at least one control parameter.

Hence, the signal processing is performed on segments of the information signal. For example, when the information signal represents a signal as a function of time, such as an audio signal, a video signal, or the like, the signal processing may be performed on a frame basis. Hence, the signal is divided into a sequence of segments, the so-called frames, the individual frames are processed, and the processed frames are combined into a resulting processed signal. In other embodiments, e.g. where the information signal is a function of a spatial parameter, a similar segmenting may be performed, e.g. by dividing a signal representing image pixels into a sequence of rows of pixels, or the like.

Consequently, by comparing the processed signal segments with the corresponding segments of the information signal, the control parameter(s) controlling the signal processing may be adjusted for each segment individually.

It is understood that the first and second segments of the information signal may have the same or different lengths.

The adjusted parameters may be applied in the processing of a subsequent segment, preferably the next segment in the sequence of segments. Hence, in this case, the second segment of the information signal is a segment subsequent to the first segment of the information signal in the sequence of segments of the information signal, thereby providing a particularly simple implementation of a self-calibrating control mechanism.

In another preferred embodiment, the second segment of the information signal is the first segment of the information signal, the first segment of the information signal being delayed to compensate for a duration of the steps of comparing the processed signal with the information signal and of adjusting the at least one control parameter. Hence, according to this two-stage feed-forward embodiment, the segment which was the basis for the adjustment of the control parameters is delayed, and the signal processing based on the

5

adjusted control parameters is applied to the first signal segment. Hence, the adjustment already takes affect for the current signal segment rather than delaying the effect of the adjustment to the next segment.

Consequently, the relative delay between the adjustment of the control parameters and the changes in the quality of the resulting processed signal is removed, thereby further improving the perceptual quality of processed signal. It is a further advantage of this embodiment that it avoids possible instabilities of the control feedback, e.g. in the case of non-stationary signals, such as audio, thereby further improving the quality of the signal processing.

According to another preferred embodiment, the method further comprises
– delaying the information signal to compensate for a duration of the steps of comparing the processed signal with the information signal and of adjusting the at least one control parameter; and

– applying at least a part of the signal modification process to the delayed information signal resulting in a modified processed signal, the at least part of the signal modification process being controlled by the adjusted at least one control parameter.

Hence, by applying at least part of the signal modification process using the adjusted control parameters on the delayed information signal, the relative delay between the adjustment of the control parameters and the changes in the quality of the resulting processed signal is removed, thereby further improving the perceptual quality of processed signal.

The at least part of the signal modification process may comprise the entire signal modification process performed based on the adjusted parameters. Alternatively, it may comprise one or more sub-processes of the signal modification process, e.g. a mixing or embedding stage, while re-using the result(s) of one or more other sub-processes, such as a watermark generation stage, which is not affected by the control parameters. In this situation, a more efficient processing may be implemented using fewer components.

As mentioned above, according to one embodiment of the invention, the processing of the information signal comprises embedding a watermark in the information signal, where the watermark is generated according to a watermark embedding model, preferably comprising a psycho-acoustic model of the human auditory system.

Accordingly, in a preferred embodiment of the invention, the signal modification process comprises
– determining a watermark signal according to a watermark embedding model;
– embedding the determined watermark signal in the information signal.

6

The watermark embedding model and/or the embedding may be controlled by one or more control parameters that may be adjusted according to the invention, thereby improving the quality of the embedding process.

It is an advantage of the invention that it provides a watermark embedding
5   with a uniform perceptual quality irrespective of a particular characteristic of the host signal, i.e. the signal into which the watermark is embedded.

According to a further preferred embodiment, the signal modification process comprises

   – determining a watermark signal according to a watermark embedding model;
10   – embedding the determined watermark signal in the information signal;

   – wherein the step of embedding the determined watermark signal is controlled by the at least one control parameter; and wherein the step of applying at least a part of the signal modification process to the information signal comprises

   – delaying the information signal resulting in a delayed signal; and
15   – embedding the determined watermark signal in the delayed signal, the embedding being controlled by the adjusted at least one control parameter.

Hence, according to this embodiment a two stage feed-forward process is employed as described above, where the embedding sub-process is controlled by the control parameters, thereby providing an efficient and stable embedding process resulting in an
20   information signal having a high and uniform perceptual quality.

According to another preferred embodiment of the invention, the information signal is an audio signal and the signal modification process comprises an audio coding process.

Hence, an audio coding method is provided which results in a coded audio
25   signal having a high and uniform quality.

According to a further preferred embodiment, the information signal is an audio signal and the signal modification process comprises an audio coding process comprising

   – determining a bit-allocation pattern for coding audio signal; and
30   – performing a quantization of the audio signal according to the determined bit-allocation resulting in a quantized signal;

wherein the step of comparing the processed signal with the information signal comprises

   – reconstructing the audio signal from the quantized signal; and

7

— comparing the reconstructed signal with the audio signal;

wherein the step of adjusting said at least one control parameter comprises adjusting the bit-allocation;

and wherein the step of applying at least a part of the signal modification process to the
5   information signal comprises

— delaying the audio signal resulting in a delayed signal; and

— performing a quantization of the delayed signal according to the adjusted bit-allocation
       resulting in a processed quantized signal.

Hence, an efficient and stable two-stage feed-forward coding process is
10   provided.

It is noted that the features of the method described above and in the following
may be implemented in software and carried out in a data processing system or other
processing means caused by the execution of computer-executable instructions. The
instructions may be program code means loaded in a memory, such as a RAM, from a storage
15   medium or from another computer via a computer network. Alternatively, the described
features may be implemented by hardwired circuitry instead of software or in combination
with software.

The present invention can be implemented in different ways including the
method described above and in the following, an arrangement, and further product means,
20   each yielding one or more of the benefits and advantages described in connection with the
first-mentioned method, and each having one or more preferred embodiments corresponding
to the preferred embodiments described in connection with the first-mentioned method and
disclosed in the dependant claims.

The invention further relates to an arrangement for processing an information
25   signal, the arrangement comprising:

— signal processing means for applying a signal modification process to an information
       signal resulting in a processed signal, said signal modification process being controlled
       by at least one control parameter;

— means for comparing the processed signal with the information signal to determine a
30   measure of perceptual quality of the processed signal; and

— means for adjusting said at least one control parameter in response to the determined
       measure of perceptual quality.

Here, the terms signal processing means, means for comparing the processed
signal with the information signal, and means for adjusting the control parameters comprise

8

suitable circuits and/or devices suitably adapted to perform the above functions. In particular, the above terms comprise general- or special-purpose programmable microprocessors, Digital Signal Processors (DSP), Application Specific Integrated Circuits (ASIC), Programmable Logic Arrays (PLA), Field Programmable Gate Arrays (FPGA), special purpose electronic circuits, etc., or a combination thereof.

The invention further relates to a device comprising an arrangement as described above and in the following.

The invention further relates to a processed information signal generated by performing the steps of the method described above and in the following.

The processed information signal may be embodied as a signal on a carrier wave, e.g. an electromagnetic wave. The information signal may further be embodied on a storage medium.

Examples of storage media include magnetic tape, optical disc, digital video disk (DVD), compact disc (CD or CD-ROM), mini-disc, hard disk, floppy disk, ferro-electric memory, electrically erasable programmable read only memory (EEPROM), flash memory, EPROM, read only memory (ROM), static random access memory (SRAM), dynamic random access memory (DRAM), synchronous dynamic random access memory (SDRAM), ferromagnetic memory, optical storage, charge coupled devices, smart cards, PCMCIA card, etc.

These and other aspects of the invention will be apparent and elucidated from the embodiments described in the following with reference to the drawing in which:

Fig. 1 shows a schematic block diagram of an arrangement for signal processing according to an embodiment of the invention;

Figs. 2a-b show schematic block diagrams of examples of an arrangement for signal processing according to another embodiment of the invention;

Fig. 3 shows a block diagram of an arrangement for embedding a watermark in an information signal according to an embodiment of the invention;

Fig. 4 shows a block diagram of an arrangement for embedding a watermark in an information signal according to another embodiment of the invention; and

Fig. 5 shows a block diagram of an arrangement for audio coding according to an embodiment of the invention.

9

It is noted that, in the figures, dashed lines between blocks generally indicate control lines while full lines between boxes generally indicate data lines.

5   Fig. 1 shows a schematic block diagram of an arrangement for signal processing according to an embodiment of the invention. The arrangement comprises a signal processing unit 101 that receives an information signal x, applies a signal modification process to the information signal x resulting in an output signal y, and outputs the signal y. The signal modification of the signal processing unit is controlled by one or more control

10   parameters generated by a quality-assessing unit 102 and fed into the signal processing unit. The information signal x and the output signal y are fed into the quality-assessing unit 102. The quality assessing unit 102 compares the processed signal y with the information signal x and determines the perceptual quality of the processed signal compared to the information signal x, i.e. a measure of the extent to which a recipient of the information signal would

15   perceive a distortion or other quality degradation of the processed signal y. The perceptual quality of an information signal may be quantified by one or more parameters, for example non-linear distortions, harmonic distortions, modulation differences, a mask-to-noise ratio, loudness asymmetry of an audio signal, etc.

Based on the determined perceptual quality of the processed signal y, the

20   quality-assessing unit generates a control signal 103 which represents the control parameter(s) for controlling the signal processing unit. Hence, an on-line self-calibrating signal processing is provided.

Figs. 2a-b show schematic block diagrams of arrangements for signal processing according to another embodiment of the invention.

25   The arrangement according to fig. 2a is implemented as a two-stage feed-forward circuit comprising a first signal processing unit 201, a quality-assessing unit 202, and a second signal processing unit 203. The first signal processing unit 201 receives the information signal x and generates a corresponding processed signal 204. The signal processing unit 201 is controlled by a default set of control parameters p. The processed

30   signal is fed into the quality-assessing unit 202 which further receives the information signal x and generates a control signal 206 based on a determination of the perceptual quality of the processed signal 204, as described above.

According to this embodiment, the control signal generated by the quality-assessing unit 202 is fed into the second signal processing unit 203. The second signal

processing unit 203 also receives the information signal x and applies to it the same signal
modification process as the first signal processing unit 201. However, the second signal
processing unit 203 is controlled by the control signal 206 instead of the default parameters p.
The second signal processing unit thus applies an optimized modification process with

5   respect to the perceptual quality of the resulting processed signal. The processed signal y
generated by the second signal processing unit thus constitutes the output of the arrangement.

It is noted that, in one embodiment, the default parameters p may be
dynamically selected rather than statically, e.g. based on the output of the quality-assessing
unit 202 or by some other control mechanism.

10   The arrangement according to fig. 2b is implemented as a two-stage feed-
forward circuit comprising two different signal processing units 201 and 207. The first signal
processing unit 201 performs the signal processing on the information signal x as described
above, and the resulting processed signal is forwarded to the quality-assessing unit 202 which
generates a control signal 206 based on a determination of the perceptual quality of the

15   processed signal 204.

According to this embodiment, the second signal processing unit 207 only
performs a part of the signal processing performed by the signal processing unit 201. Hence,
according to this embodiment, the first signal processing unit 201 forwards a signal 205 to
the second signal processing unit 207 where the signal 205 represents the result of a first

20   processing stage which is not influenced by the control parameters p and 206, respectively.
The second signal processing unit 207 only performs a second part of the signal processing
controlled by the optimized parameters 206 and resulting in the final processed signal y. It is
an advantage of this embodiment, that unnecessary repetitions of signal processing steps are
voided, thereby reducing the processing time and the required complexity of the components.

25   It is understood that the above embodiments may comprise additional
components, such as delay circuits for compensating for delays introduced by the various
components.

Fig. 3 shows a block diagram of an arrangement for embedding a watermark
in an information signal according to an embodiment of the invention. The arrangement

30   comprises a watermark embedder 301 that receives an audio signal x, embeds a watermark in
the audio signal, and generates a resulting watermarked audio signal y. The watermark
embedder 301 comprises a watermark generation module 302 that receives the audio signal x
and generates a watermark signal 304 on the basis of a watermark w representing the
information to be embedded. The watermark generation module 302 is controlled by an

11

embedding model block 305 that receives the audio signal x, determines a predetermined cost function derived from the audio signal x and generates a control signal 306 controlling the watermark generation module 302. For example, the control signal may comprise parameters controlling one or more properties of the watermark generation process, such as the relative

5   watermark energy embedded at different frequencies, window sizes for the segmenting of the signal, etc. The watermark embedder further comprises a mixing module 303 that receives the audio signal x and the watermark signal 304 generated by the watermark generation module 302. The mixing module combines the audio signal x and the watermark signal 304 resulting in the watermarked signal y. The embedding model 305 further generates a control

10 signal 313 which is fed into the mixing module 303 and which controls the mixing of the watermark signal and the audio signal. For example, the embedding model may determine the amount of watermark energy to be added to the audio signal based on the properties of the audio signal, e.g. the amount of noise that can be masked by the signal. This headroom is generally referred to as mask-to-noise ratio.

15   According to the invention, the audio signal x is delayed by a delay 314 to compensate for the delay introduced by the embedder 301. The delayed audio signal 315 and the watermarked signal y are fed into a quality-assessing unit 307. The quality-assessing unit 307 compares the watermarked signal y and the delayed audio signal 315 and passes a corresponding set of quality parameters 309 to a control unit 308. For example, the quality

20 assessing unit 307 may implement a cost function that quantifies the artifacts of the watermark system or a more complex system, e.g. as described in Thilo Thiede et al., "PEAQ - The ITU Standard for Objective Measurement of Perceived Quality", J. Audio Eng. Soc., Vol. 48, No. 1/2, 2000. The quality parameters 309 quantify the perceptual quality of the watermarked audio signal y. Examples of such quality parameters include non-linear

25 distortion, harmonic distortion, modulation difference, mask to noise ratio, loudness asymmetry, etc.

The control unit 308 converts the quality parameters 309 into appropriate control parameters for controlling the settings of the embedder 301. The control unit feeds control signals 310, 311, and 312 to the mixing module 303, the watermark generation

30 module 302, and the embedding model 305, respectively. Each of the control signals comprises one or more of the control parameters generated by the control unit. For example, the control parameters may control embedding parameters such as the watermark embedding strength, the frame size, the time/frequency allocation, etc., or combinations of the above parameters. For example, if the measured quality is low, the settings of the embedding model

and/or the watermark generation module and/or the mixing module are adjusted in a direction that improves the quality of the watermarked signal. On the other hand, if the quality is sufficiently high to allow a stronger embedding, the settings are adjusted as to allow more watermark energy to be embedded in the audio signal, thereby improving the robustness of
5   the embedded watermark.

The above conversion of the quality parameters into control parameters may comprise a simple scaling, other simple conversion functions such as a mixing matrix, and/or more complex conversions, such as neural networks, statistical methods, or the like.

Hence, in this embodiment, the adjusted control parameters are used to
10   determine the embedder settings for subsequent parts of the audio signal. For example, in a frame-based embedding system where the audio signal is divided into frames and the watermark is embedded in each of the frames, the control parameters adjusted based on the quality of a given frame of the watermarked signal are applied to the embedding of the watermark of the subsequent frame.

15   It is noted that in an alternative embodiment, the embedding model only controls one of the watermark generation module and the mixing module. Similarly, in alternative embodiments, the control unit may only forward control parameters to one or two of the modules of the watermark embedder.

Fig. 4 shows a block diagram of an arrangement for embedding a watermark
20   in an information signal according to another embodiment of the invention. The arrangement according to this embodiment implements a two-stage feed-forward mechanism.

The arrangement comprises a watermark embedder 401 comprising an embedding model 405, a watermark generation module 402, and a mixing module 403. The embedder 401 embeds a watermark w in an audio signal x resulting in a watermarked signal
25   y' as described in connection with the embedder 301 in fig. 3 above. The watermarked signal y' is fed into the quality-assessing unit 407 which compares the watermarked signal y' with the delayed audio signal 415 generated by an appropriate delay 414 from the audio signal x. The quality-assessing unit 407 generates a set of quality parameters 409 and feeds them into a control unit 408 which, in turn, generates a set of control parameters 410, as was described
30   in connection with the quality-assessing unit 307 and the control unit 308 of fig. 3 above.

According to this embodiment, the control unit does not control the embedder 401 but a separate mixing module 421. The mixing module 421 receives the delayed audio signal 420 which is suitably delayed by delays 414 and 425 in order to compensate for the delays introduced by the embedder 401 and the quality-assessing unit 407 and the control

unit 408. The mixing module 421 further receives a delayed watermark signal 423 corresponding to the watermark signal 404 generated by the watermark generation module 402 and delayed by a delay 424. The delay 424 compensates for the delay introduced by the quality-assessing unit 407 and the control unit 408, and the delayed watermark signal 423 is

5   fed into the mixing module 421. The mixing module 421 further receives the control signal 413 generated by the embedding model 405 for controlling the mixing process. Again, the control signal is delayed by the delay 424 resulting in a delayed control signal 422. The mixing module combines the delayed audio signal 420 with the delayed watermark signal 423 according to the control parameters 410 and the delayed control signal 422 from the

10 embedding model, resulting in the watermarked signal y that constitutes the output of the arrangement.

Hence, according to this embodiment, the mixing stage of the embedding process is repeated where the mixing parameters of the repeated stage are set by the control circuit according to the perceptual parameters determined after the initial mixing.

15 It is an advantage of this embodiment that it provides an improved control of the embedding system, since it removes the relative delay between the controlling parameters and the controlled system. Consequently, the perceptual quality of the watermarked signal is further improved, and the risk of a non-stable feedback system is eliminated.

It is noted that such a feed-forward system is equivalent to a one-iteration

20 feedback system where the delay between the control parameters and the signal itself is compensated.

Hence, in the above embodiments, self-calibrating watermark embedding systems are disclosed which provide a uniform quality of the watermarked signal. The strength of the watermark and/or the setting of the embedding algorithm are continuously

25 adjusted using a quality-assessing unit so as to maintain a uniform audio quality.

Fig. 5 shows a block diagram of an arrangement for audio coding according to an embodiment of the invention. The arrangement comprises an audio coder 501 for coding an audio signal x. The audio coder comprises an encoder model 505, e.g. based on a psycho-acoustic model of the human auditory system. The encoder model 505 generates a bit-

30 allocation pattern 506 which is fed into a quantization module 502. The audio coder further comprises a reconstruction module 503 that reconstructs the audio signal from the output of the quantization module 502. It is understood that the reconstruction module also receives the bit allocation pattern from the encoder model 505. The reconstructed audio signal x' is fed into a quality-assessing unit 507 which compares the original audio signal, suitably delayed

14

by the delay circuit 514, with the reconstructed audio signal x' and generates corresponding quality parameters 509 which are fed into a control unit 508. The control unit converts the quality parameters into quantization control parameters 510. The quantization control parameters are fed into a second quantization module 521. The second quantization module

5   receives the audio signal x after suitable delay by the delays 514 and 525. The second quantization module further receives the output 506 from the encoder model 505 after suitable delay by delay 524. The second quantization module 521 performs an improved quantization with an improved bit allocation based on the control parameters received from the control unit 508. Finally, the output 529 of the second quantization module is encoded in

10   the encoder 530 resulting in a bit-stream signal y.

Hence, in the above, a self-calibrating audio encoder is disclosed.

It is noted that the above arrangements may be implemented as general- or special-purpose programmable microprocessors, Digital Signal Processors (DSP), Application Specific Integrated Circuits (ASIC), Programmable Logic Arrays (PLA), Field Programmable Gate

15   Arrays (FPGA), special purpose electronic circuits, etc., or a combination thereof.

It should be noted that the above-mentioned embodiments illustrate rather than limit the invention, and that those skilled in the art will be able to design many alternative embodiments without departing from the scope of the appended claims.

For example, in the above, the invention has mainly been described in

20   connection with audio signals. It is understood, however, that it may also be applied to other information signals, such as video signals, multimedia signals, images, etc.

The invention can be summarized as follows. Disclosed is a method of processing an information signal comprising: applying a signal modification process to an information signal resulting in a processed signal, said signal modification process being

25   controlled by at least one control parameter; comparing the processed signal with the information signal to determine a measure of perceptual quality of the processed signal; and adjusting said at least one control parameter in response to the determined measure of perceptual quality. The method is advantageously used in watermarking and compression systems, that use a model of the Human Auditory or Visual System to control the watermark

30   embedding depth or the compression ratio, and that often fail to maintain a constant signal quality because they ignore other artifacts than the mask-to-noise ratio.

In the claims, any reference signs placed between parentheses shall not be construed as limiting the claim. The word "comprising" does not exclude the presence of

15

elements or steps other than those listed in a claim. The word "a" or "an" preceding an element does not exclude the presence of a plurality of such elements.

The invention can be implemented by means of hardware comprising several distinct elements, and by means of a suitably programmed computer. In the device claim enumerating several means, several of these means can be embodied by one and the same item of hardware. The mere fact that certain measures are recited in mutually different dependent claims does not indicate that a combination of these measures cannot be used to advantage.